

WORLD



TELECOM LABS



SIP Support V2.0

Recent Upgrades – Feb 06

- Digest authentication (extra security for SIP phones etc)
- 'Keep Alive' message added (had caused problems with Quintum gateways)
- Support for embedded SIP Proxy (similar to H323 GNU Gatekeeper support)

SIP Support

- SIP available for IPNx and PVx (Solaris R/M only)
- SIP supported by IPNx AG & DG gateways
- Remote upgrade available for installed product
- Same capacity limits as H323
- SIP NOT supported by INx & Desktop PVx

Features

- Voice calls
- DTMF supported
 - RFC 2833 method
 - Cisco proprietary 'INFO' method
- Fax calls
 - G711 fallback
 - IPNx AG proprietary
- IPNx is a SIP user agent

Key Benefits

- NOP for SIP will save bandwidth
- TDM, VoIP, SS7, Calling Card all in same box
- H323, SIP & NOP all co-exist
- Any-protocol-in, Any-protocol-out design for maximum flexibility
- IPNx & SoIP Gateway give SIP to SS7 support
- Soft IVR is a SIP application server

Key Benefits 2

- SIP 'Digest' authentication allows direct connect SIP users (Broadband telephony applications)
- All calls (VoIP or TDM) treated the same in IPNx & PVx
- ... therefore inherit all applications (Pre-Paid, Callback, LCR, CDR generation etc)
- SIP trace facility built-in

SIP User Agent

- SIP has 2 elements: User Agents & Registrar
- SIP agent requires 3 functions to build a SIP gateway:
 - client side of the registrar (how to register in a SIP network)
 - call control (how to handle call)
 - SDP manager: the codec and bandwidth negotiation
- WTL supports hosted proxy and registrar if required

SIP Proxy & Registrar Support

- SIP Express Router (SER) can be run within PVx or IPNx
- Independent (not integrated) service
- Most SIP Proxy functions already handled by switch software
- Proxy main benefit is routing calls to SIP direct connect customers

RFCs Supported

- RFC 2833 for DTMF transmission
- RFC 2327 (Session Description Protocol)
- RFC 3261 (Session Initiation Protocol)
- RFC 2543 User Agent
- RFC 2617 Digest Authentication

Compatibility

- Successful interconnect testing done with range of:
 - SIP Gateways
 - Handsets
 - PC Clients
- However, variations still exist in SIP implementations. Contact WTL to test into our SIP Interconnect Server for any new device.

Compatibility 2

- SIP Handsets, Gateways & ATAs
 - Sipura SPA-841 phone, Sipura SPA-1000 and SPA-2000 ATA
 - Draytek router with built-in sip phone (DrayTek UA-1.1)
 - Accel gateway with sip stack
 - Asterisk PBX PA168S
 - Centrality PA168S IP Phone
 - Artdio ip phone
 - Sippy (CiscoSystemsSIP-GW-UserAgent 5591)
 - Quintum ASG400, Quintum AXG800
 - Mediarings Voicbridge Session Border Controller

Compatibility 3

- PC Clients/ Softphone
 - x-ten softphone dialler
 - helmsman dialler
 - SIPPs, Ahead
 - x-lite pc dialer (X-Lite Free World Dialup build 1082)
 - Express Talk v1.03
- Carriers
 - Verscom, Turkey

Compatibility 4

- Note following features not currently supported:
 - TCP Signalling (DTMF transport method)
 - ‘Registrar’ function (unless SER installed)
- Also note Fax & DTMF support may be problematic with new 3rd party gateways

Typical Uses

- Residential/business gateways
- Direct connect of subscribers
- SIP feature server model
- Carrier interconnect